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20. ABSTRACT (Continue on reverse side if necessary and identify by block number)

A modified Recursive Least-Square algorithm is presented for echo cancellation. The modified RLS algorithm freeze the adaptive gain during double-talking periods to improve the convergence performance of adaptive echo canceller.

A lattice filter structure of echo canceller also given based on exact least-squares algorithm. The new algorithm with superior convergent speed can be implemented using VLSI.

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APPLICATION OF LEAST-SQUARES ALGORITHMS TO ADAPTIVE ECHO CANCELLATION+

V. U. Reddy, F. K. Soong, A. M. Peterson and T. Kailath*

SUMMARY

With the advent of commercial communication satellites, roundtrip delays of the order of 500 ms in long distance telephone conversations have become quite common. Use of echo suppressors on such long distance telephone circuits has not given satisfactory performance and hence attention has been directed to the use of adaptive echo cancellation[1,2].

Essentially all the significant echoes are generated at the hybrid transformer, which acts as a two-wire/four-wire interface in a long distance telephone circuit. It is the impedance mismatches introduced by the hybrid transformer that cause echoes. Thus the echo signal is a delayed and transformed version of the speech signal. If an adaptive filter, implented using an adaptive algorithm, can simulate the transformation yielding the echo, the echo can be eliminated by subtracting the simulated echo from the actual echo signal. The combination of the adaptive filter and the subtractor is known as an adaptive echo canceler. Since perfect cancellation is only possible asymptotically, it is important to know how fast the residual echo power falls. This depends on the convergence properties of the adaptive algorithm. The algorithms that have been extensively studied so far in the context of echo cancellation are all based on stochastic gradient approximations[1,2,3]. Only recently have fast Kalman estimation algorithms and lattice algorithms been applied to adaptive channel equalization[4,5]. The echo-killer chip recently developed by Bell Telephone Laboratories[6] uses the so-called least-mean-squares(LMS) algorithm for adaptation. One good feature of this algorithm is its simplicity.

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Under the conditions of one-way speech, i.e., when only one speaker is talking, it would suffice if the canceler adapts to the unknown echo path as quickly as possible and continues to track small variations in the echo path. However, in normal telephone conversations there are many intervals during which double-talking, i.e., two speakers talk simultaneously, takes place. During such periods, the second speaker's speech acts as a gigantic additive noise for the echo canceler which is trying to cancel the echoes of the first speaker. This forces the adapted path to diverge from its previously adapted values. This has two effects: i) The canceler performs very poorly during the double-talking intervals and ii) the canceler needs to re-adapt to the unknown echo path after the second speaker stops talking. Hence, it would be highly desirable to have an algorithm that not only has fast convergence but also stops the adapted path from diverging during the double-talking interval. This paper contains the results of a study with the above objectives.

Two different structures are considered for the echo canceler. One uses the usual tapped-delay-line(TDL) filter and the other is based on the lattice configuration. The TDL canceler was simulated using two different adaptive algorithms: i) LMS(a stochastic gradient version) and ii) recursive least-squares(RLS). The lattice-form canceler was simulated using the corresponding exact least-squares algorithm. A typical hybrid transformer, which forms the unknown echo path, and the two test speakers' signals were simulated on the lines suggested in the standard CCITT report[7]. Fig. 1 shows the impulse response of the hybrid transformer.

A TDL echo canceler is shown in Fig.2, where x_t is the far-end speech signal(first speaker's signal), y_t is its echo and, v_t is the sum of the near-end speech signal(second speaker's signal) and the white noise n_t (system noise). The RLS update of the coefficient vector $A^T=(a_0, a_1, \ldots, a_L)$ is given by

$$A_t = A_{t-1} + \frac{P_{t-1} X_t \, (z_t - X_t^T A_t)}{1 + X_t^T P_{t-1} X_t} \qquad (1)$$

$$P_t = P_{t-1} - \frac{P_{t-1} X_t X_t^T P_{t-1}}{1 + X_t^T P_{t-1} X_t} \qquad (2)$$

where $X_t^T = (x_t, x_{t-1}, \ldots\ldots, x_{t-L})$ and $z_t = y_t + v_t$. For the lattice-form

echo canceler and the corresponding least-squares lattice (LSLAT) algorithm, see[8].

Figures 3a and 3b illustrate the convergence behavior of the 31-tap TDL and lattice-form echo cancelers, respectively. The plots show that the RLS and LSLAT algorithms converge in about 100 iterations and the residual error power at convergence is practically equal to -40 dB which is the power of the system noise n_t. The plots also show that the RLS and LSLAT algorithms behave identically after a small number of iterations, say 50. The difference during the initial period is due to the different initial conditions assumed in the two algorithms.

For the purpose of illustration, the TDL canceler was also simulated using the LMS algorithm. The constants of the algorithm were adjusted to give the best possible performance. Figure 4 gives the learning curve. The results show that the LMS algorithm gives a residual error power that, even after 800 iterations, essentially converges to a level about 6 dB above the system noise level.

To overcome the double-talking(two-way speech) problem, a modification of the RLS algorithm is proposed. The modified algorithm is given by

$$A_t = A_{t-1} + \frac{P_{t-1} X_t (z_t - X_t^T A_t)}{1/g_t + X_t^T P_{t-1} X_t} \tag{3}$$

$$P_t = P_{t-1} - \frac{P_{t-1} X_t X_t^T P_{t-1}}{1/g_t + X_t^T P_{t-1} X_t} \tag{4}$$

where

$$g_t = Var(n_t)/ Var(v_t) \tag{5}$$

(For the corresponding modified LSLAT algorithm , see[8]).

During the double-talking interval $1/g_t$ assumes a large value, i.e., of the order of 10^3. This forces the second term in (3) and (4) to be very small. How small these terms become depends on the relative values of P_t and P_{t-1}. It is known that P_t tends to zero at the rate of 1/t. Since there is always a time lag between the starting of the conversation (at which time the adaptive algorithm begins) and the initial double-talking, the term $X_t^T P_{t-1} X_t$ is effectively negligible compared to $1/g_t$ at the instant the double-talking starts. Thus, the modified algorithm of (3) and (4)

virtually freezes F_t and A_t at the instant the double-talking begins and continues to do so for the whole double-talking interval.

To verify the performance of the modified algorithm, an echo path consisting of a flat delay of 20 samples in cascade with the hybrid transformer, giving an impulse response that is a 20-sample delayed version of the one shown in Fig.1, and a 51-tap adaptive filter were chosen for the simulation. Only the lattice-form canceler was used in the experiment. The normal LSLAT algorithm was run initially. The modified version was started at 401-th iteration and the second speaker's signal was started at 501-th iteration. The algorithm was run for a total number of 1000 iterations and the impulse response of the adapted path was computed from the filter parameters at 1000-th iteration. In the un-modified case, the normal algorithm was run for 1000 iterations.

Figures 5a and 5b show the impulse response of the adapted path without and with modification. Comparing these results with Fig.1, it is clear that the modified algorithm leaves the adapted path almost unaffected by the double-talking. On the other hand, the adapted path is severly distorted by the double-talking when the modification is not used.

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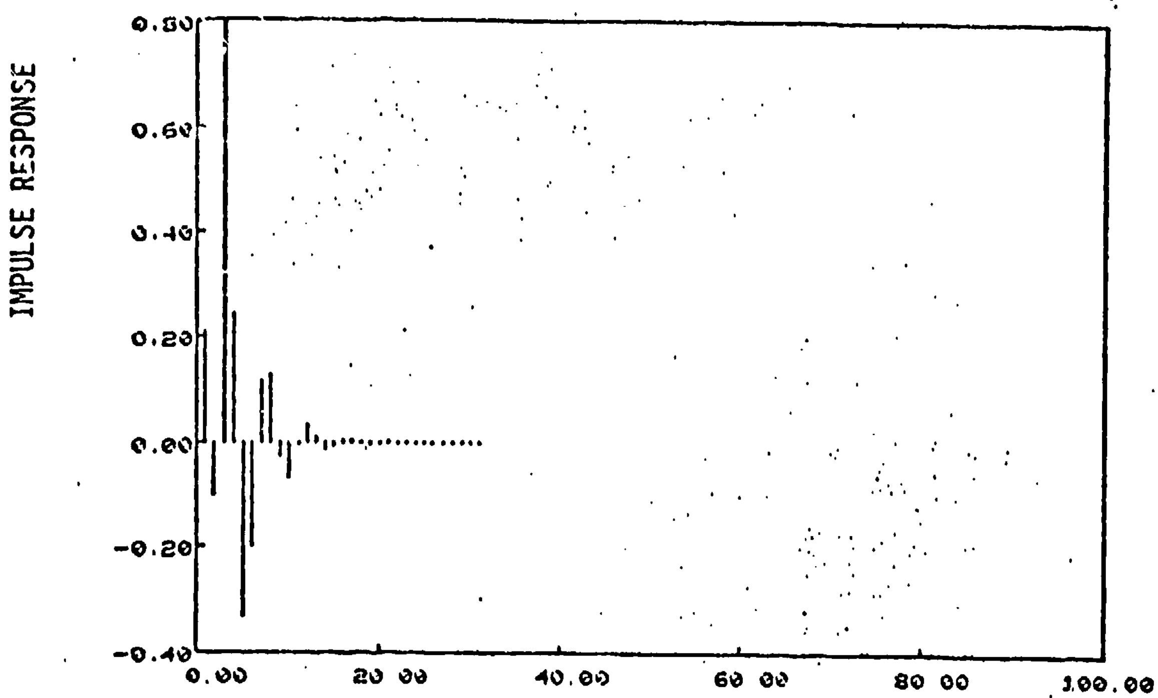


Fig. 1 Impulse response of the simulated hybrid transformer

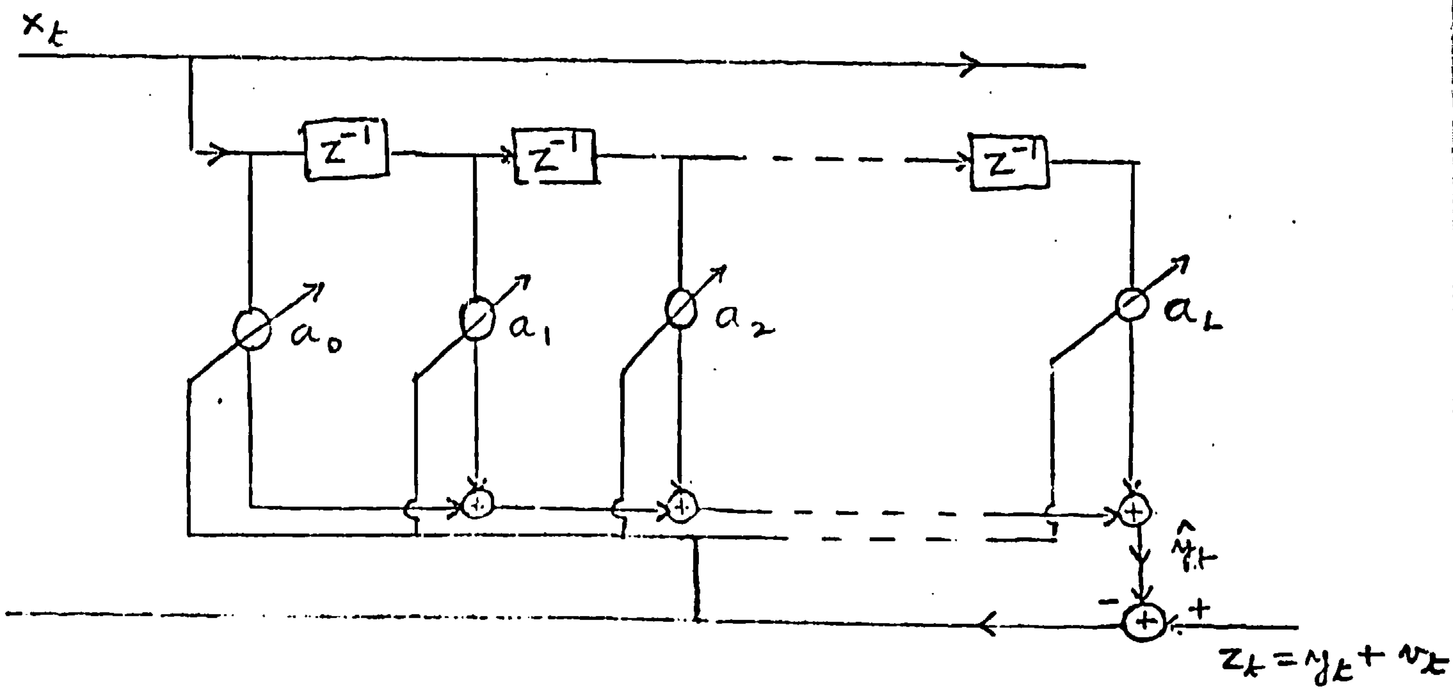
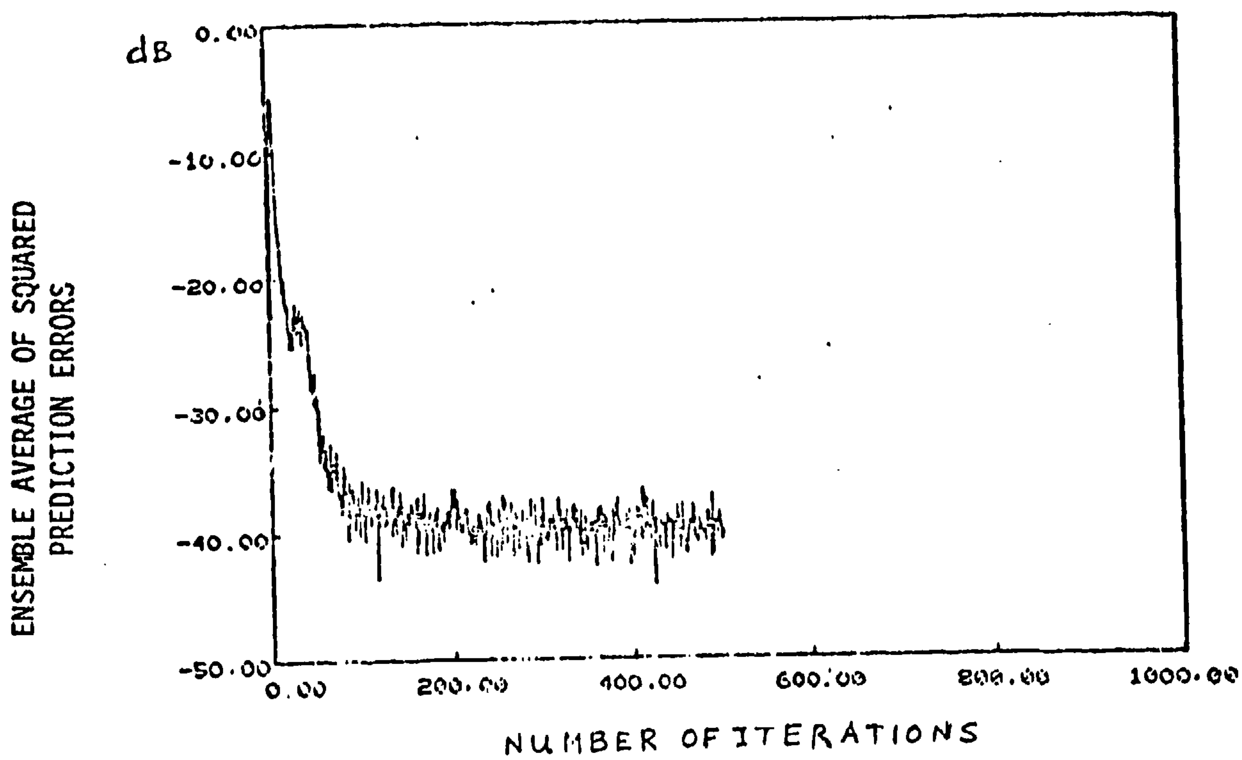
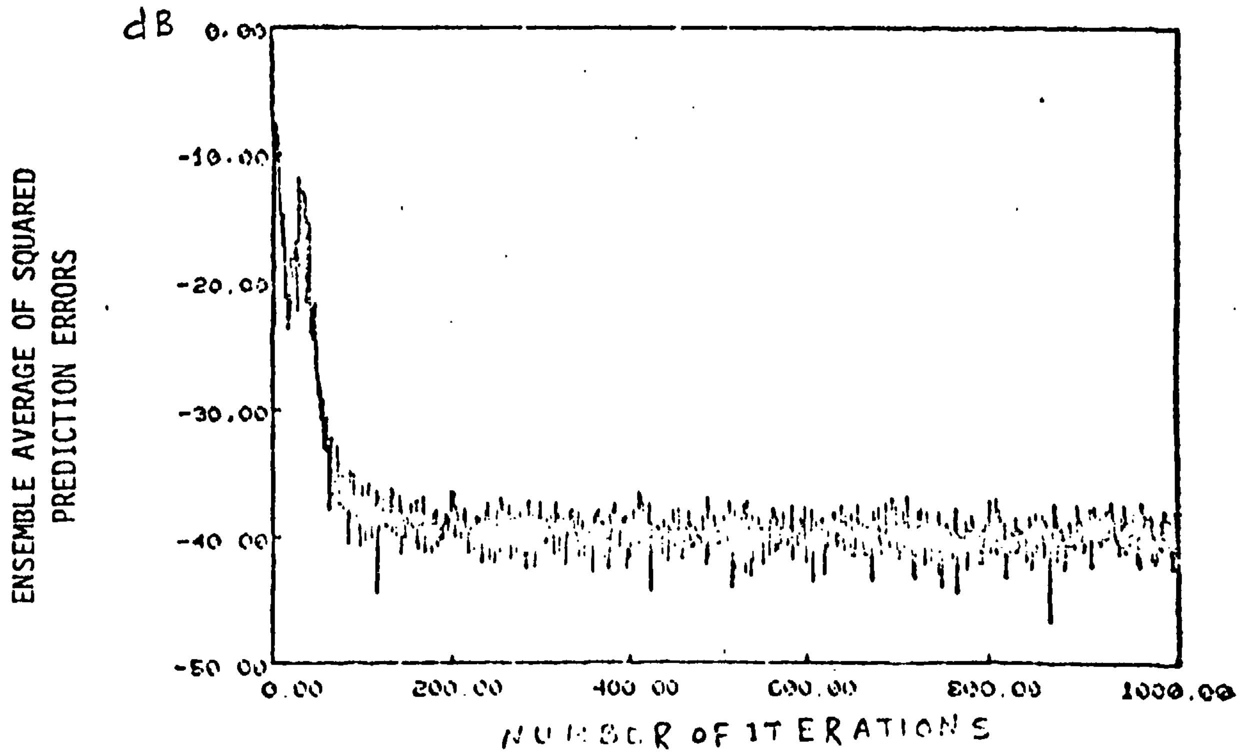


Fig.2 TDL echo canceler



a. RLS algorithm



b. LSLAT algorithm

Fig. 3 Convergence behavior of the two algorithms

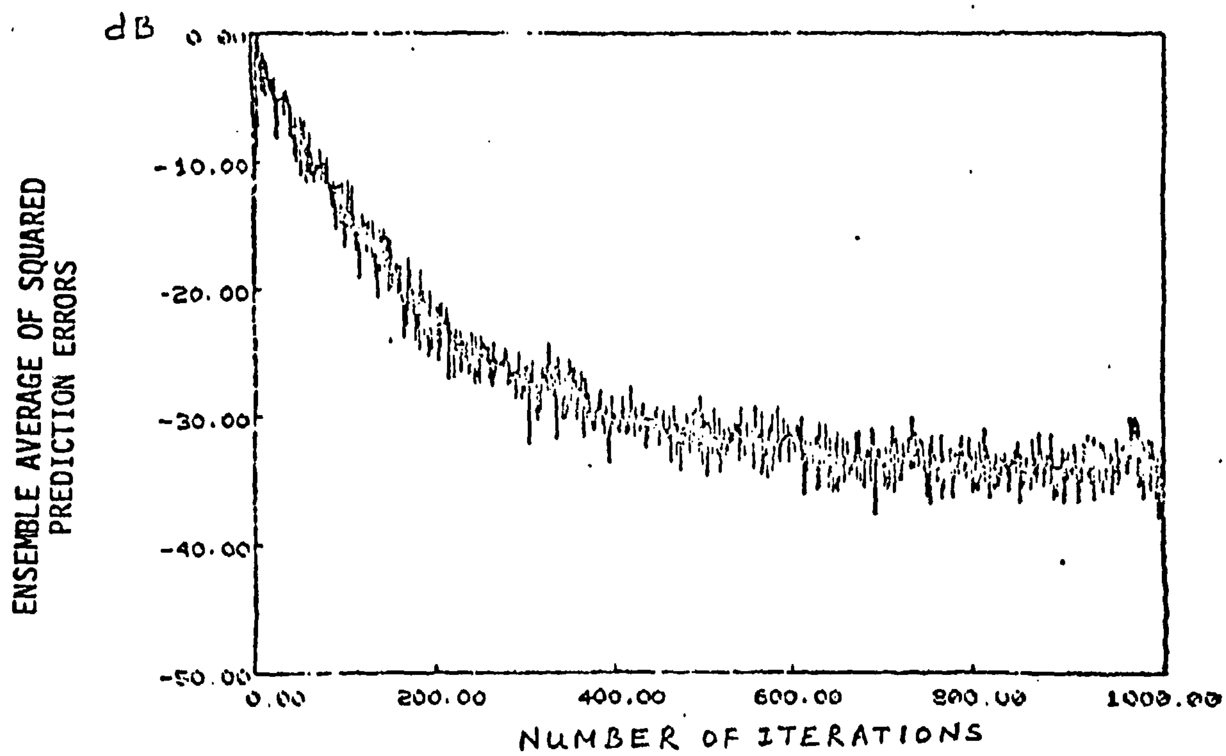
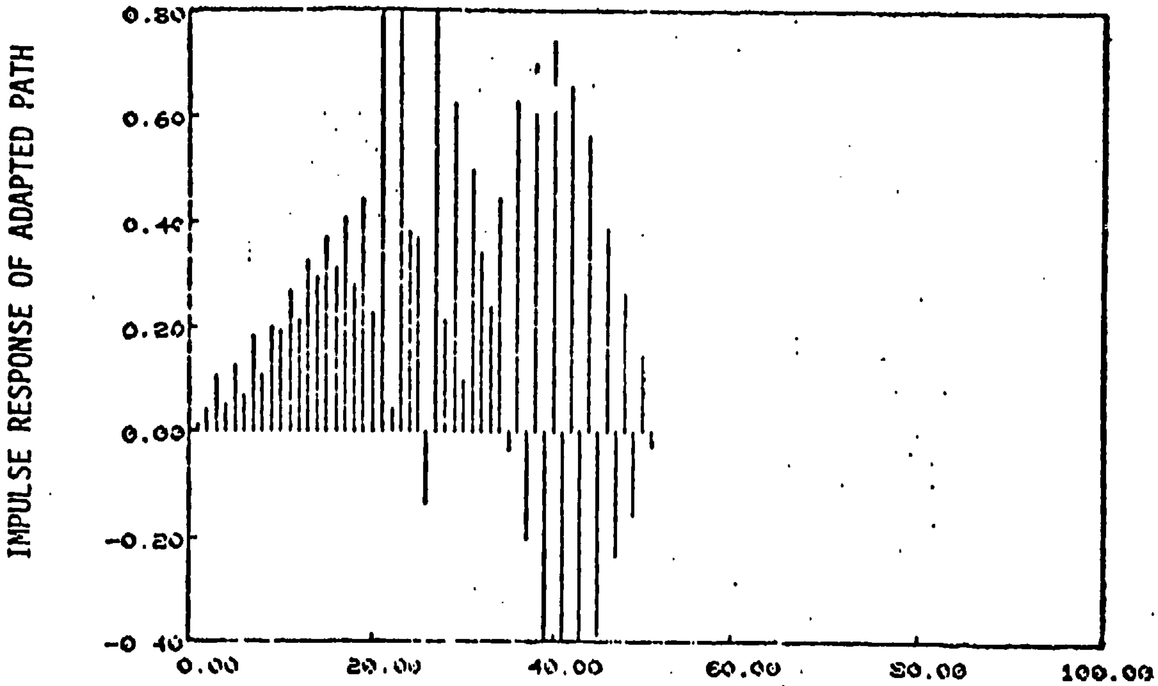
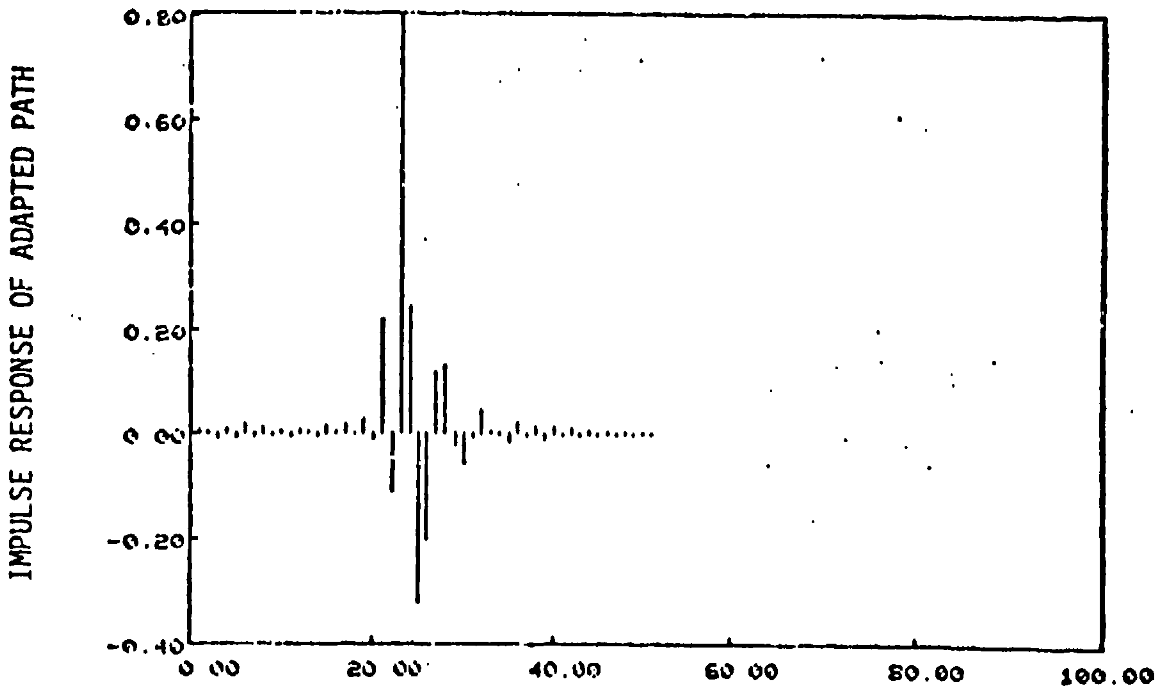


Fig. 4 Convergence behavior of the LMS algorithm



a. Without modification



b. With modification

Fig. 5 Impulse response of the adapted path using the LSLAT algorithm